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Carvalho de Matos

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(54) **SYSTEM AND METHOD FOR MEASURING SOUND**

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(51) **Int. Cl.**

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G10L 25/00	(2013.01)
H04R 3/00	(2006.01)
H04R 5/027	(2006.01)

(52) **U.S. Cl.**

CPC ***H04R 3/005*** (2013.01); ***H04R 5/027*** (2013.01)

(58) **Field of Classification Search**

CPC G10L 21/0208; G10L 25/78; G10L 25/48; G10L 21/00; G10L 21/02; G10L 21/0232; G10L 21/0364; G10L 25/03; H04N 21/42203; G06F 3/16; H04R 3/005; A63F 13/215

USPC 704/226, 271, E21.003, 205, 206, 268, 704/270, 220, 276, 275; 381/59, 94.1; 702/75, 85; 73/646; 600/559, 300

See application file for complete search history.

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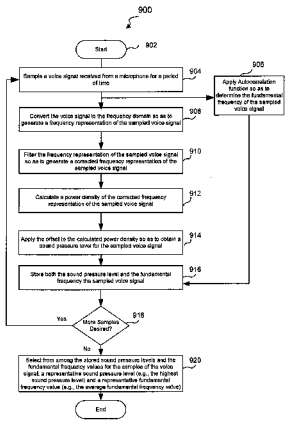
(57) **ABSTRACT**

A system and method for measuring sound is described. In one embodiment frequency-banded-noise samples, which collectively cover at least a portion of a spectrum, are sequentially generated at different points in time, and a baseline sound-pressure-level reading for each of the frequency banded noise samples is received. Using data received from a microphone, a sound pressure level reading is generated for each of the frequency banded noise samples. Calibration data is then produced for the microphone as a function of a difference between each of the baseline sound-pressure-level readings and a corresponding one of each of the generated sound pressure level readings for each of the frequency banded noise samples.

22 Claims, 9 Drawing Sheets



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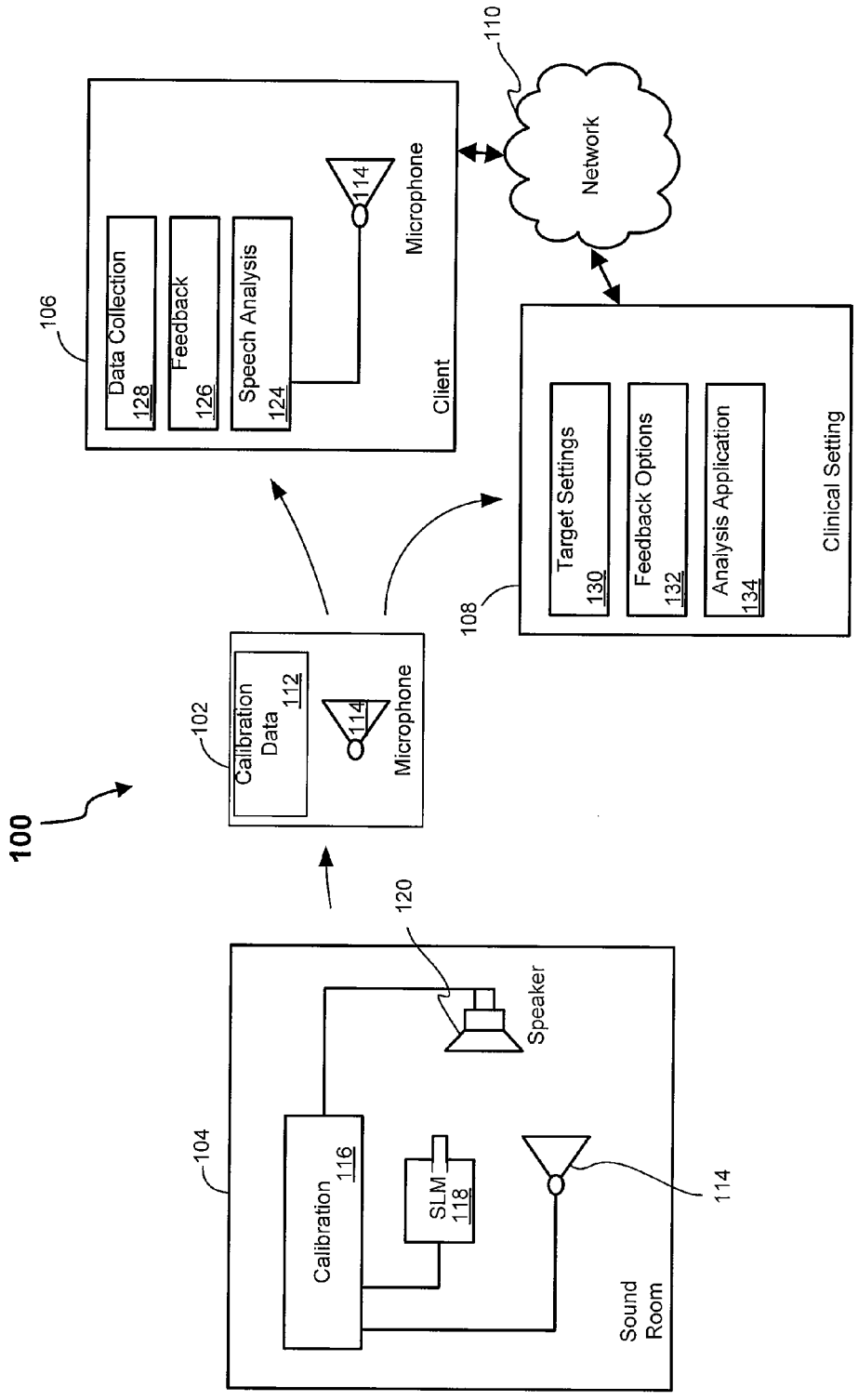
100
102
Calibration Data 112
114
Microphone
104
Calibration 116
SLM 118
Sound Room
120
Speaker
114
106
Data Collection 128
Feedback 126
Speech Analysis 124
114
Microphone
Client
110
Network
108
Target Settings 130
Feedback Options 132
Analysis Application 134
Clinical Setting

FIGURE 1

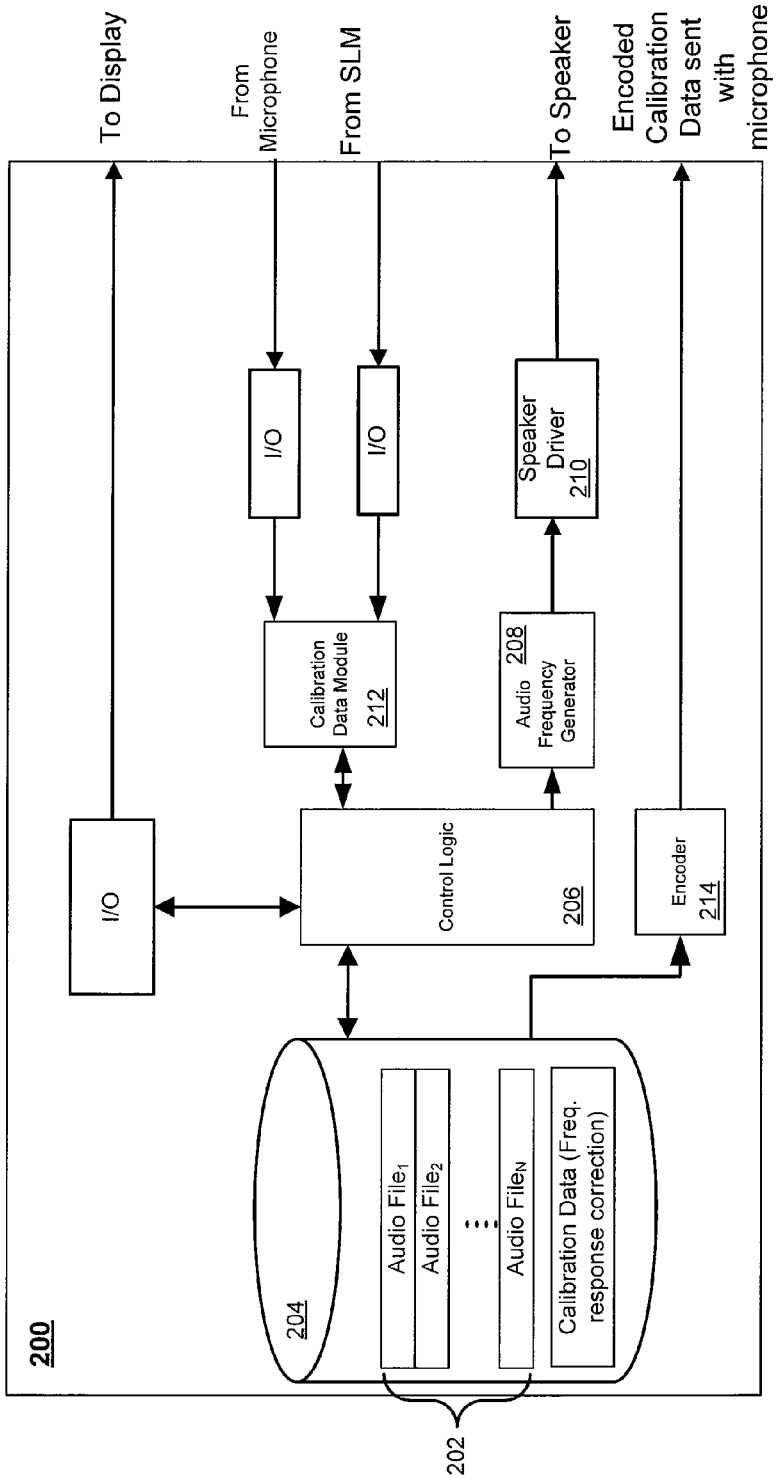
To Display
From Microphone
From SLM
To Speaker
Encoded Calibration Data sent with microphone
200
I/O
I/O
I/O
Speaker Driver 210
Calibration Data Module 212
Audio Frequency Generator 208
Control Logic 206
Encoder 214
204
Audio File1
Audio File2
Audio FileN
Calibration Data (Freq. response correction)
202

FIGURE 2

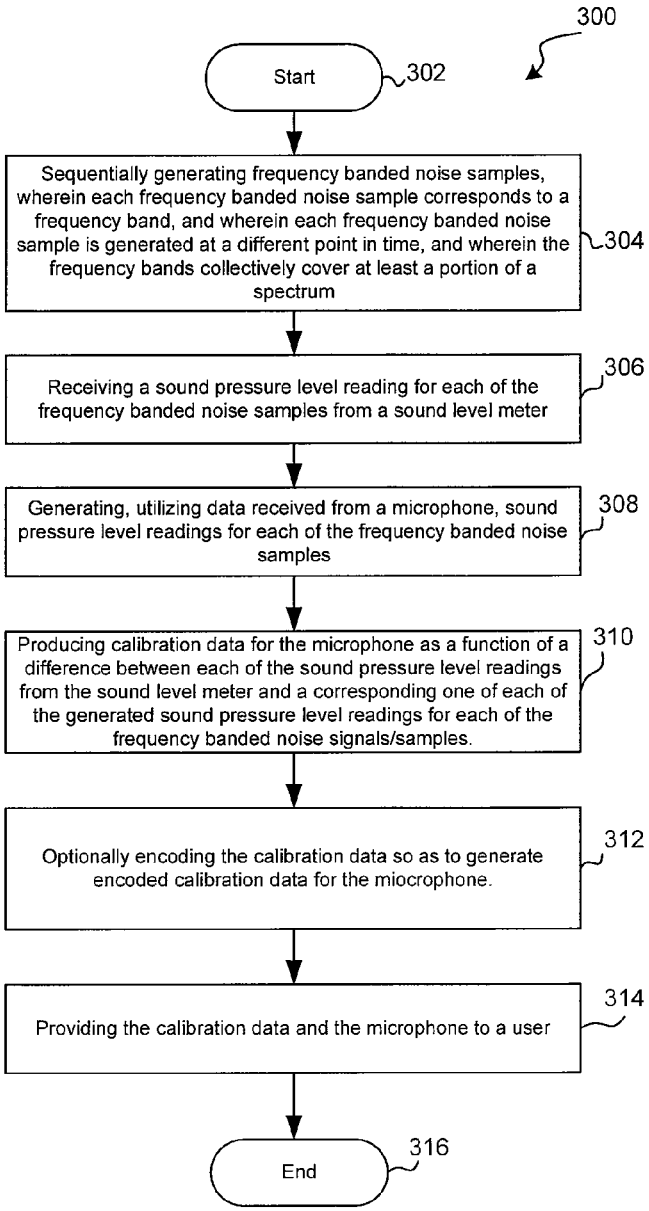
300
Start
302
Sequentially generating frequency banded noise samples, wherein each frequency banded noise sample corresponds to a frequency band, and wherein each frequency banded noise sample is generated at a different point in time, and wherein the frequency bands collectively cover at least a portion of a spectrum
304
Receiving a sound pressure level reading for each of the frequency banded noise samples from a sound level meter
306
Generating, utilizing data received from a microphone, sound pressure level readings for each of the frequency banded noise samples
308
Producing calibration data for the microphone as a function of a difference between each of the sound pressure level readings from the sound level meter and a corresponding one of each of the generated sound pressure level readings for each of the frequency banded noise signals/samples.
310
Optionally encoding the calibration data so as to generate encoded calibration data for the miocrophone.
312
Providing the calibration data and the microphone to a user
314
End
316

FIGURE 3

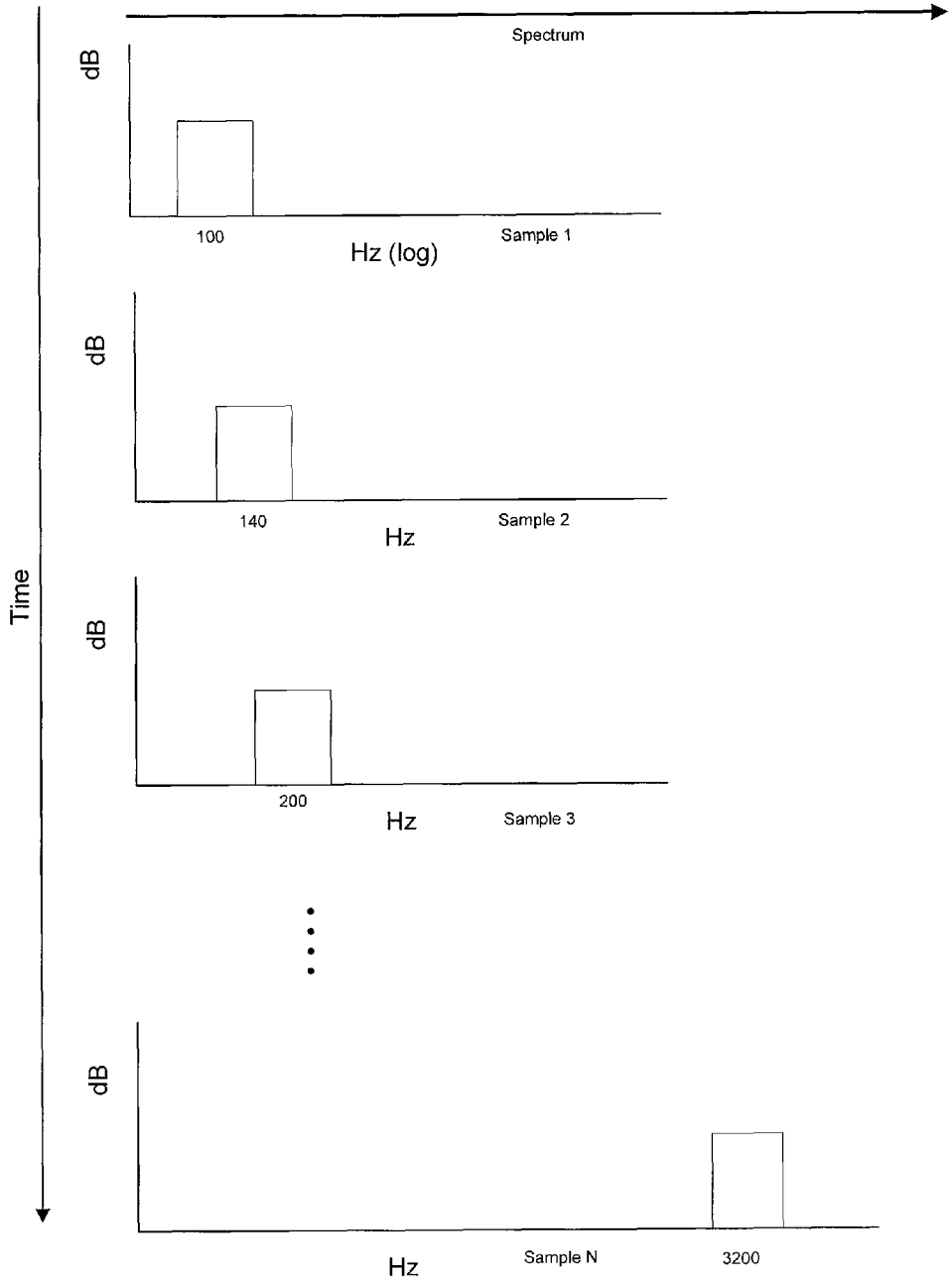
Spectrum
dB
100
Hz (log)
Sample 1
dB
140
Hz
Sample 2
Time
dB
200
Hz
Sample 3
dB
Hz
Sample N
3200

FIGURE 4

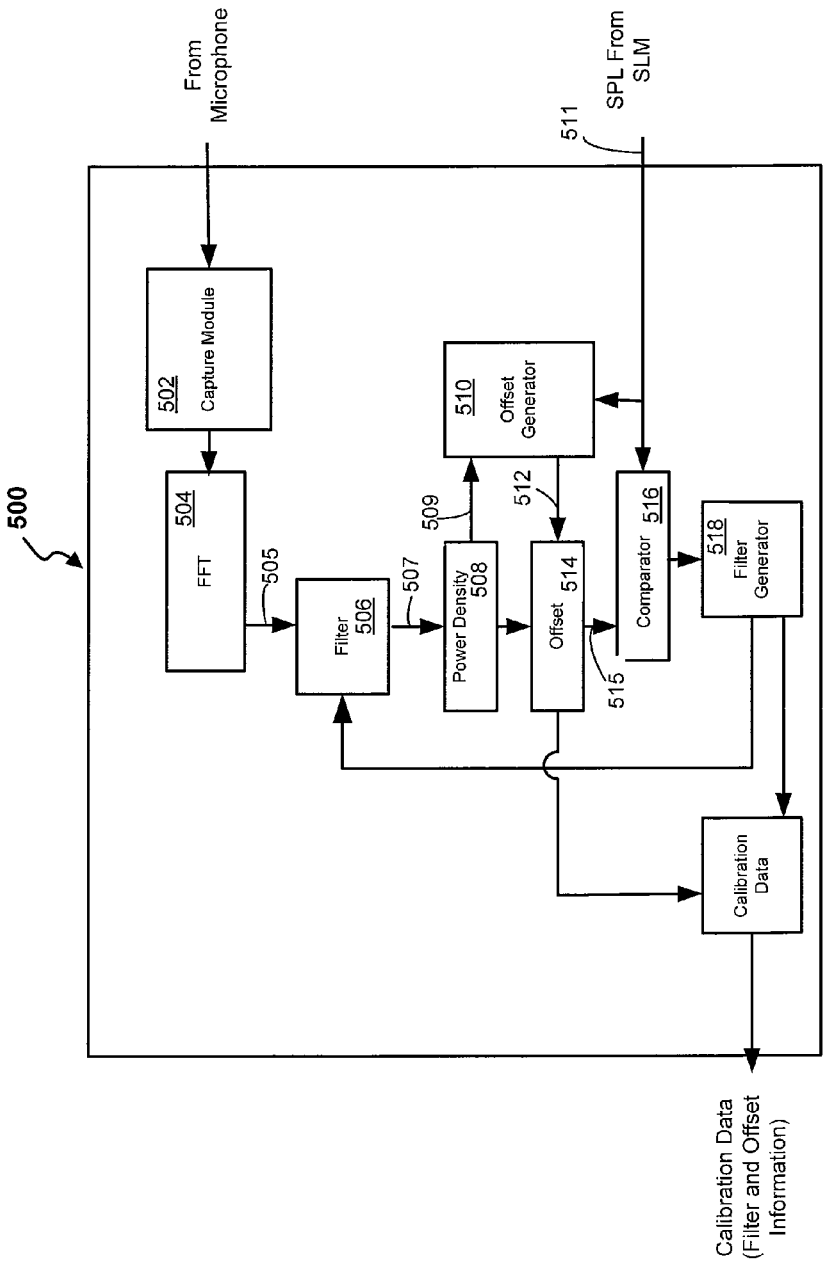
500
From Microphone
SPL From SLM
511
502 Capture Module
504 FFT
505
Filter 506
507
Power Density 508
509
510 Offset Generator
512
Offset 514
515
Comparator 516
518 Filter Generator
Calibration Data
Calibration Data (Filter and Offset Information)

FIGURE 5

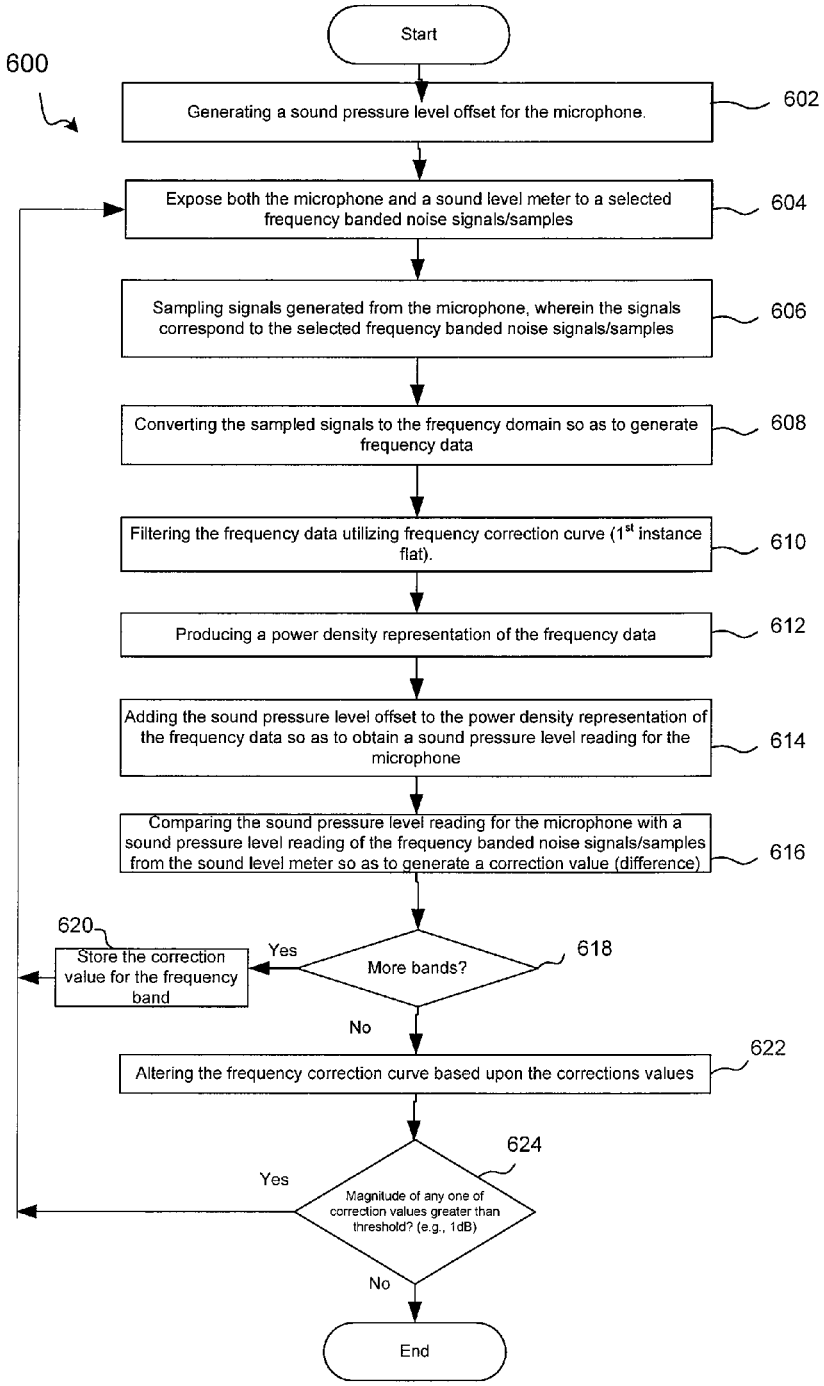
600
Start
Generating a sound pressure level offset for the microphone.
602
Expose both the microphone and a sound level meter to a selected frequency banded noise signals/samples
604
Sampling signals generated from the microphone, wherein the signals correspond to the selected frequency banded noise signals/samples
606
Converting the sampled signals to the frequency domain so as to generate frequency data
608
Filtering the frequency data utilizing frequency correction curve (1st instance flat).
610
Producing a power density representation of the frequency data
612
Adding the sound pressure level offset to the power density representation of the frequency data so as to obtain a sound pressure level reading for the microphone
614
Comparing the sound pressure level reading for the microphone with a sound pressure level reading of the frequency banded noise signals/samples from the sound level meter so as to generate a correction value (difference)
616
More bands?
618
Yes
620
Store the correction value for the frequency band
No
Altering the frequency correction curve based upon the corrections values
622
Magnitude of any one of correction values greater than threshold? (e.g., 1dB)
624
Yes
No
End

FIGURE 6

```
////////////////////////////////////////////////////////////////
SPL AutoCalibrator (v0.15c): 2005-7-13 18:46 // AK5370_002 50cm side-below
Checking noise floor...
4(±1.1): SLM = 56.7dB; MIC = -73.6
Adjusting volume to 75dB...
Volume: -20
Volume: -22
Choosing microphone gain...
Microphone gain set to 1 (minimum)
Testing SPL ramp...
4(±0.1): SLM = 60.8dB; MIC = -57.8
4(±0.1): SLM = 63.3dB; MIC = -54.9
4(±0.1): SLM = 66.3dB; MIC = -51.9
3(±0.1): SLM = 69.1dB; MIC = -48.9
4(±0.1): SLM = 72.1dB; MIC = -45.9
4(±0.1): SLM = 75.1dB; MIC = -43.0
4(±0.1): SLM = 78.0dB; MIC = -40.0
4(±0.1): SLM = 81.1dB; MIC = -37.0
4(±0.1): SLM = 84.0dB; MIC = -33.9
4(±0.1): SLM = 87.0dB; MIC = -30.9
4(±0.1): SLM = 90.0dB; MIC = -28.0
Phase 1 complete:
Initial offset = 118.0
MIC SPL ceiling > 90dB
SPL error for SLM is 0.3dB (OK)
SPL error for MIC is 0.1dB (OK)
Testing frequency response...
4(±0.6): FRC[ 100] +=  7.8 =  7.8
5(±0.4): FRC[ 140] +=  7.7 =  7.7
5(±0.4): FRC[ 200] +=  5.4 =  5.4
4(±0.2): FRC[ 280] +=  1.6 =  1.6
4(±0.2): FRC[ 400] += -1.9 = -1.9
4(±0.1): FRC[ 560] += -4.9 = -4.9
5(±0.3): FRC[ 800] +=  0.7 =  0.7
4(±0.3): FRC[1120] += -2.0 = -2.0
4(±0.1): FRC[1600] += -1.1 = -1.1
4(±0.3): FRC[2240] +=  1.0 =  1.0
4(±0.2): FRC[3200] +=  1.6 =  1.6
Max FRC error = 7.8
4(±0.6): FRC[ 100] += -0.0 =  7.8
4(±0.3): FRC[ 140] +=  0.1 =  7.8
5(±0.4): FRC[ 200] +=  0.4 =  5.8
3(±0.0): FRC[ 280] += -0.2 =  1.5
4(±0.3): FRC[ 400] +=  0.1 = -1.8
5(±0.2): FRC[ 560] += -1.0 = -5.9
4(±0.4): FRC[ 800] +=  1.0 =  1.7
5(±0.2): FRC[1120] += -0.2 = -2.2
4(±0.1): FRC[1600] +=  0.1 = -1.0
4(±0.2): FRC[2240] +=  0.4 =  1.5
4(±0.2): FRC[3200] +=  0.2 =  1.8
Max FRC error = 1.0
4(±0.5): FRC[ 100] +=  0.2 =  8.0
4(±0.4): FRC[ 140] +=  0.0 =  7.8
4(±0.4): FRC[ 200] +=  0.2 =  6.1
5(±0.2): FRC[ 280] +=  0.1 =  1.6
4(±0.2): FRC[ 400] +=  0.3 = -1.6
4(±0.3): FRC[ 560] += -0.2 = -6.1
3(±0.4): FRC[ 800] +=  0.4 =  2.0
3(±0.2): FRC[1120] += -0.2 = -2.4
5(±0.1): FRC[1600] +=  0.0 = -1.0
4(±0.3): FRC[2240] +=  0.0 =  1.5
3(±0.2): FRC[3200] += -0.0 =  1.7
Max FRC error = 0.4 (OK)
Performing final correction...
4(±0.1): SLM = 75.1dB; MIC = 75.1; diff = -0.1 (OK)
Final offset = 118.0
SPL floor = 40.8dB
Calibration complete and looks good!
```

FIGURE 7

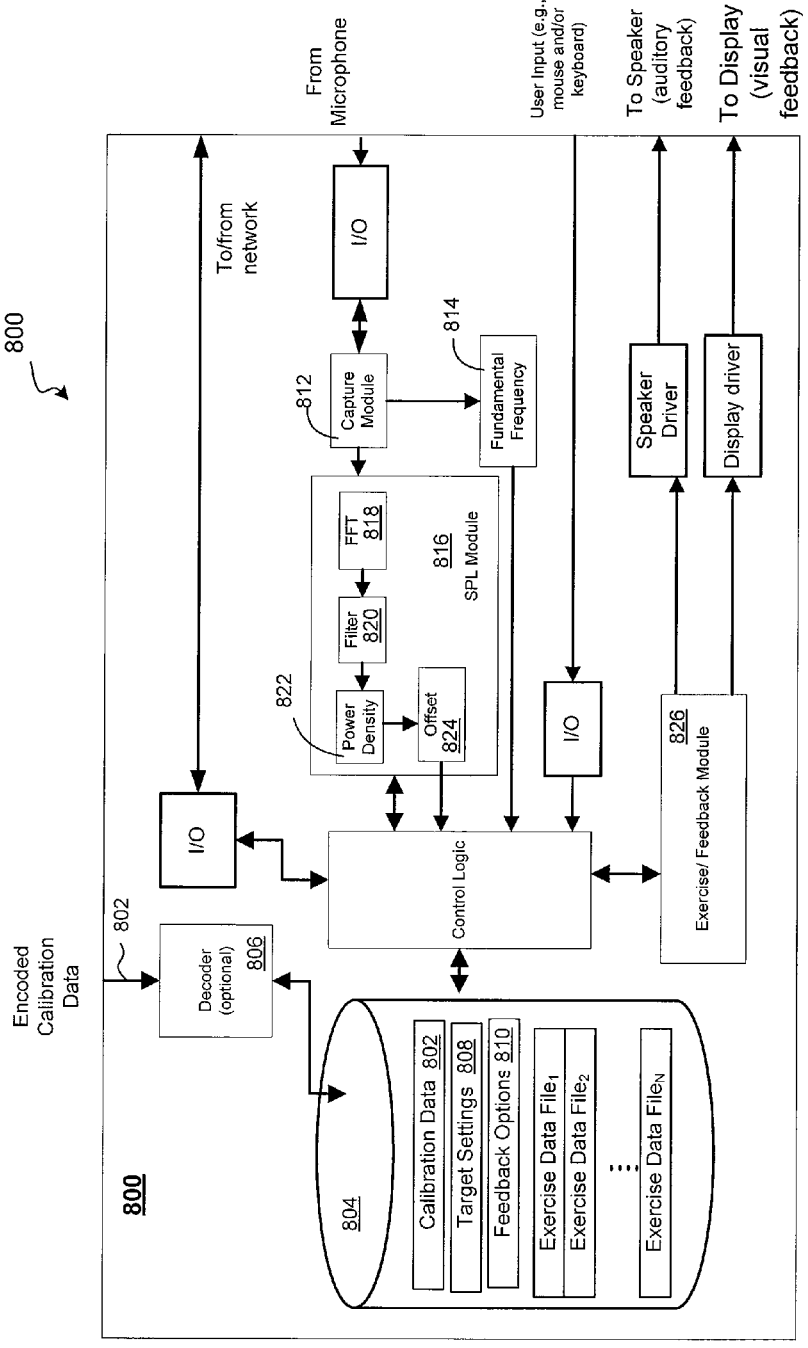
800
Encoded Calibration Data
802
Decoder (optional) 806
I/O
To/from network
From Microphone
I/O
812
Capture Module
814
Fundamental Frequency
FFT 818
Filter 820
Power Density
822
Offset 824
816 SPL Module
User Input (e.g., mouse and/or keyboard)
I/O
Control Logic
Speaker Driver
Display driver
To Speaker (auditory feedback)
To Display (visual feedback)
826
Exercise/ Feedback Module
800
804
Calibration Data 802
Target Settings 808
Feedback Options 810
Exercise Data File1
Exercise Data File2
Exercise Data FileN

FIGURE 8

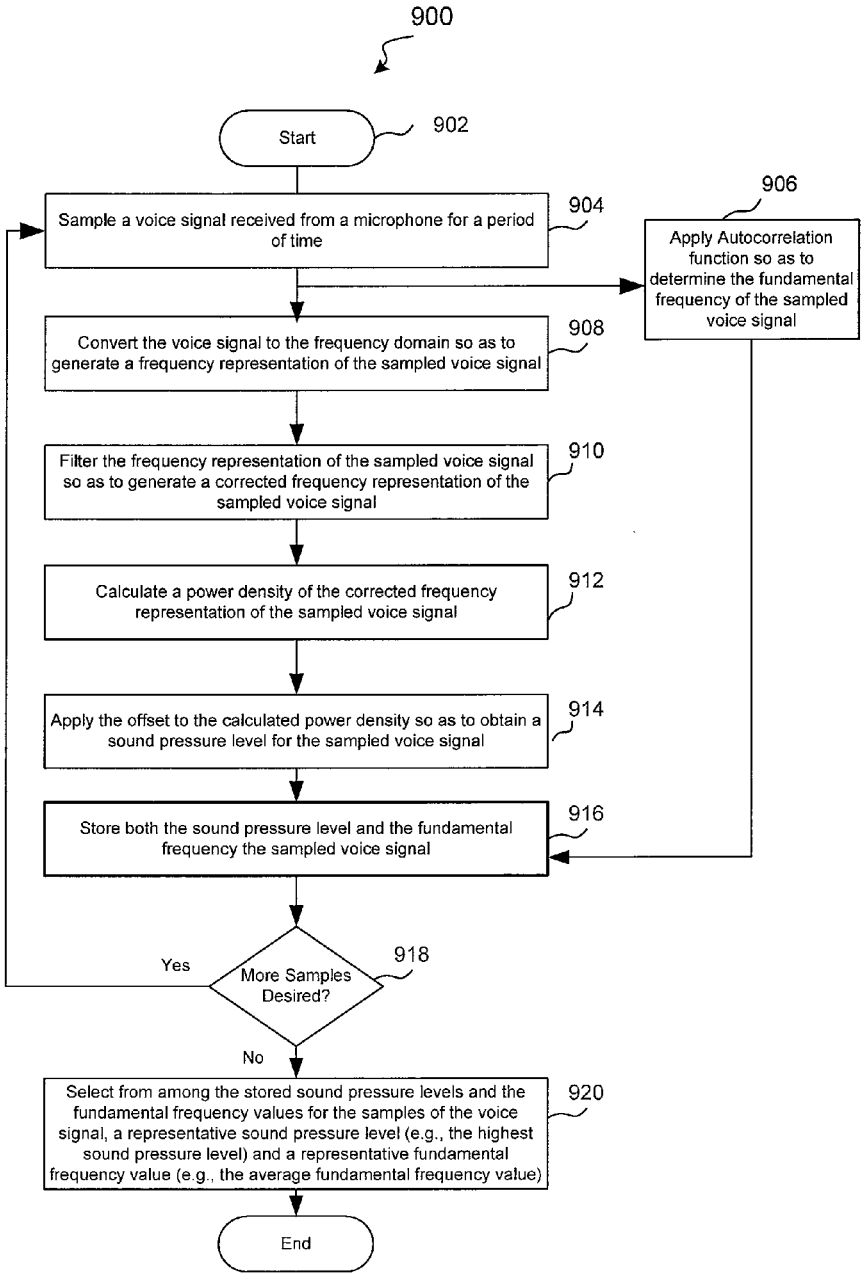
900
Start
902
Sample a voice signal received from a microphone for a period of time
904
906
Apply Autocorrelation function so as to determine the fundamental frequency of the sampled voice signal
Convert the voice signal to the frequency domain so as to generate a frequency representation of the sampled voice signal
908
Filter the frequency representation of the sampled voice signal so as to generate a corrected frequency representation of the sampled voice signal
910
Calculate a power density of the corrected frequency representation of the sampled voice signal
912
Apply the offset to the calculated power density so as to obtain a sound pressure level for the sampled voice signal
914
Store both the sound pressure level and the fundamental frequency the sampled voice signal
916
Yes
More Samples Desired?
918
No
Select from among the stored sound pressure levels and the fundamental frequency values for the samples of the voice signal, a representative sound pressure level (e.g., the highest sound pressure level) and a representative fundamental frequency value (e.g., the average fundamental frequency value)
920
End

FIGURE 9

SYSTEM AND METHOD FOR MEASURING SOUND

PRIORITY

The present application claims priority to commonly owned and assigned application No. 60/714,005, filed Sep. 2, 2005, entitled System and Method for Measuring Sound, which is incorporated herein by reference.

FIELD OF THE INVENTION

This invention relates generally to sound measurement, and more particularly to systems and methods for improved sound measurement.

BACKGROUND

There are several known disorders that adversely affect the speech of many people. For example, Parkinson disease (PD) multiple sclerosis, strokes, ataxic dysarthria, aging voice, and vocal fold paralysis impact the speech of many persons. Traditional neuropharmacological and neurosurgical treatments have been of limited help in improving these problems with conventional wisdom being that speech disorders are resistant to medical treatments and efforts at traditional speech therapy have been considered ineffective.

The landscape, however, has changed recently, and experimental data from a focused program of specialized speech therapy has been shown to provide significant benefits, which may include improved speech intelligibility, motor functions and neural functioning. These treatment effects have been shown to be relatively long lasting without additional treatment and have been considered the first Type 1 evidence for speech treatment for PD.

Speech treatment is an immediate, practical, and relatively inexpensive intervention for improving behavior. There is no requirement for FDA regulation, and an efficacious treatment is available, easily delivered and highly acceptable to patients with minimal, if any, negative side effects.

Speech treatment may be effectively carried out in a clinical setting with sophisticated hardware and speech analysis equipment. But these clinical locations may be inconvenient, unavailable and/or too expensive for many patients to take advantage of. Although microphones and rudimentary speech analysis software are available, typical consumer-grade microphones are currently not able to accurately measure one or more important characteristics of the user's voice. For example, measurements of the sound pressure level and fundamental frequency of a user's voice are often too inaccurate with many affordable microphones to provide the type of analysis desired for speech therapy purposes.

SUMMARY OF THE INVENTION

Exemplary embodiments of the present invention that are shown in the drawings are summarized below. These and other embodiments are more fully described in the Detailed Description section. It is to be understood, however, that there is no intention to limit the invention to the forms described in this Summary of the Invention or in the Detailed Description. One skilled in the art can recognize that there are numerous modifications, equivalents and alternative constructions that fall within the spirit and scope of the invention as expressed in the claims.

The present invention can provide a system and method for measuring sound. In one embodiment, frequency-banded-noise samples, which collectively cover at least a portion of a spectrum, are sequentially generated at different points in time, and a baseline sound-pressure-level reading for each of the frequency banded noise samples is received. Using data received from a microphone, a sound pressure level reading is generated for each of the frequency banded noise samples. Calibration data is then produced for the microphone as a function of a difference between each of the baseline sound-pressure-level readings and a corresponding one of each of the generated sound pressure level readings for each of the frequency banded noise samples.

DETAILED DESCRIPTION

Referring now to the drawings, where like or similar elements are designated with identical reference numerals throughout the several views, and referring to FIG. **1**, shown is a block diagram depicting an exemplary environment **100** in which embodiments of the present invention may be implemented. As shown in the embodiment depicted in FIG. **1**, a calibration data and microphone package **102** is provided from a sound room **104** to a client location **106**. In addition, a clinic **108** is shown in communication with the client **106** via a network **110**.

In accordance with several embodiments of the present invention, a calibration procedure carried out at the sound room **104** generates calibration data **112** for the microphone **114** that enables the microphone **114** to be utilized for purposes that the microphone **114** was previously unsuitable for. As depicted in FIG. **1** the calibration data **112** is provided to the client **106** along with the microphone **114** to enable a user to accurately measure characteristics of the user's voice. In some embodiments, for example, the calibration data **112** enables the microphone **114** to be used in connection with accurate measurements of the sound pressure level and fundamental frequency of a user's voice that would not be achievable with the microphone **114** (e.g., because of its low quality) without the calibration data **112**.

Although several embodiments of the present invention are described in the context of transducing audible sound in a speech therapy setting, it should be recognized that the calibration and sound capturing procedures of the present invention are certainly not limited to these applications. For example, it is contemplated that the accurate measurements that the calibration data **112** provides may be used in a variety of applications where, for example, sound pressure level and/or fundamental frequency readings are desired.

As depicted in FIG. **1**, the sound room **104** includes a calibration module **116** coupled to a sound level meter **118**, a speaker **120** and a microphone **114**. The calibration module **116** in the exemplary embodiment is configured to calibrate the microphone **114** by generating sound that is transmitted by the speaker **120** and received by the sound level meter **118** and the microphone **114**, The sound received at the microphone **114** and sound level meter **118** is analyzed and the calibration data **112** for the microphone **114** is generated and sent with the microphone **114** to the client **106**. The calibration module **116** in some embodiments is implemented by a general purpose computing device (e.g., personal computer, laptop, PDA, cellular handset) that is configured utilizing software and/or firmware to capture and analyze the sound transmitted from the speaker **120**.

Advantageously, the calibration data **112** generated by the calibration module **116** according to several embodiments of the present invention allows inexpensive and readily available microphones to be utilized in connection with sound analysis techniques that previously required substantially more

expensive equipment. As discussed further herein, the calibration data **112** in some embodiments is encoded calibration data (e.g., to reduce its size) and in other embodiments, the calibration data **112** is raw calibration data.

The client **106** in this embodiment includes a speech-analysis unit **124**, a feedback unit **126** and a data collection unit **128**, which in several embodiments, are realized by a general purpose computing device (e.g., personal computer, laptop, PDA, cellular handset) that is configured utilizing software and/or firmware. The speech-analysis unit **124** in this embodiment is configured to receive the calibration data **112** and to operably couple with the microphone **114** so as to receive and analyze speech from a user utilizing the calibration data **112** and provide information about one or more aspects of the user's speech.

The feedback module **126** is configured to provide feedback to the user using graphical displays and/or audible feedback to facilitate proper sampling and provide a therapeutic feedback system to help improve the speech of a user. The data collection module **128** is configured to collect speech data gathered in the form of one or more files that may be transmitted to the clinic for further analysis.

The clinic **108** in this embodiment includes a target-settings module **130**, a feedback-options module **132** and an analysis module **134**. The target-settings module **130** allows a clinician to customize and provide target settings to a particular client. For example, target sound pressure levels may be established by a clinician and sent via the network **110** to the client. Similarly, the feedback-options module **132** allows a clinician to select specific forms of feedback (e.g., specific graphical interfaces) and send the selected feedback forms to the feedback module **126** at the client **106** so that the user interfaces with the speech analysis module using feedback techniques tailored to the user by the clinician.

Referring next to FIG. **2**. shown is a block diagram **200** of an exemplary embodiment of the calibration module **116** of FIG. **1**. While referring to FIG. **2**, simultaneous reference will be made to FIG. **3**, which is a flowchart depicting steps traversed by the calibration module **116**, **200** depicted in FIGS. **1** and **2**.

As shown, the calibration module **200** initially generates frequency banded noise samples at different points in time (Blocks **302**, **304**). Each of the frequency banded noise samples in this embodiment corresponds to a different frequency band, and collectively the frequency bands cover at least a portion of a spectrum (e.g., a portion of an audible sound spectrum).

Referring to FIG. **4**, for example, shown are N frequency banded noise samples that are generated sequentially one at a time in accordance with an exemplary embodiment. As shown, the noise samples are generated one at a time and each noise sample spans an octave band of a spectrum of noise (e.g., a spectrum of pink noise). In the exemplary embodiment, each successive banded noise sample overlaps a portion of the previous noise sample so that the entire spectrum is covered in a piecemeal fashion.

Referring back to the embodiment depicted in FIG. **2**, each of N audio files **202** stored in a file storage device (e.g., a hard drive) **204** corresponds to, and includes data to generate, one of N frequency banded noise samples. Control logic **206**, in connection with an audio frequency generator **208**, is configured to sequentially retrieve each of the N audio files **202** and generate the frequency banded noise signals that are transduced from electrical energy to acoustical energy by the speaker **120** in connection with the speaker driver **210**.

As shown, a sound-pressure-level reading for each of the frequency-banded-noise samples is received from the sound-level meter **118** (Block **306**), and corresponding sound-pressure-level readings are generated for each of the frequency-banded-noise samples utilizing data received from the microphone **114** (Block **308**). Next, calibration data **112** for the microphone **114** is produced by the calibration-data module **212** as a function of a difference between each of the sound-pressure-level readings from the sound level meter **118** and a corresponding one of each of the generated sound-pressure level readings for each of the frequency-banded-noise samples (Block **310**). Optionally, calibration data is encoded by an encoder **214** so as to generate encoded-calibration data for the microphone **114** (Block **312**), and then the calibration data **112** and the microphone **114** are provided to a user (Block **314**). As depicted in FIG. **1**, for example, the calibration data **112** is packaged along with the microphone **114** so that when the microphone **114** is received by a user, the user also possesses calibration data **112** that significantly improves sound readings from the microphone **114**. This is certainly not required, however, and it is contemplated that the microphone **114** and calibration data **112** may be sent to the client at different times.

Referring next to FIG. **5**, shown is a block diagram **500** depicting one embodiment of the calibration-data module **212** depicted in FIG. **2**. While referring to FIG. **5**, simultaneous reference will be made to FIG. **6**, which depicts steps traversed by the calibration-data module **500** when generating calibration data. As shown, initially a sound-pressure-level offset for the microphone **114** is generated (Block **602**).

In some embodiments, the sound-pressure-level offset is generated by capturing with a capture module **502** a noise sample (e.g., pink noise) received by the microphone **114** that is transduced by the speaker **120**. The sampled noise is then converted to the frequency domain with the Fast Fourier Transform (FFT) module **504**, and a power-density module **508** generates a power-density reading **509** from the noise represented in the frequency domain. An offset generator **510** then compares the power-density reading **509** with a sound-pressure reading **511** of the noise sample (obtained from the sound-level meter **118**) to generate an offset **512** for the microphone.

In one embodiment, noise is generated with the speaker at several power levels (e.g., every 3 dB from 60 to 90 dB) and a reading at each of the power levels is obtained with the microphone **114** and compared against the sound-pressure readings from the sound-level meter **118** so as to arrive at an accurate offset over a range of power levels.

Once the sound pressure level offset for the microphone **114** is generated, both the microphone **114** and the sound-level meter **118** are exposed to a selected frequency banded noise sample (Block **604**). In some embodiments, for example, a frequency banded pink noise sample that corresponds to a frequency band from 100 to 200 Hz is initially generated and transduced by the speaker so as to expose the microphone and sound level meter to the noise sample with frequencies from 100 to 200 Hz.

Once the microphone **114** is exposed to the frequency-banded-noise sample, signals generated from the microphone **114** are sampled by the capture module **502** (Block **606**), and converted by the FFT module **504** to the frequency domain so as to generate frequency data **505** (Block **608**). The frequency data **505** is then filtered by the filter module **506** so as to obtain filtered frequency data **507** (Block **610**). For the first frequency-banded sample, the filter module **506** employs a flat-frequency-response filter, but subsequent frequency-banded samples are filtered utilizing a response curve generated as a function of the microphone's **114** response to previous noise banded sample(s).

As shown in FIGS. **5** and **6**, after filtering, a power-density representation of the frequency data is produced by the power-density module **508** (Block **612**), and the sound pressure level offset **514** is added to the power-density representation of the frequency data so as to obtain a sound-pressure level reading **515** for the microphone (Block **614**). A comparator **516** then compares the sound-pressure reading **515** for the microphone **114** with a sound-pressure level reading of the frequency banded noise signals/sample from the sound level meter **118** so as to generate a correction value (e.g., a difference) (Block **616**).

If there are more frequency bands to sample with the microphone (Block **618**), then the correction value for the frequency band is stored and blocks **604-616** are executed again. If there are no more frequency bands to sample (Block **618**), then the frequency-correction curve is altered by the filter generator **518** based upon the correction values. In the event the magnitude of any one of correction values is greater than a threshold, (e.g., 1 dB), then data for the frequency correction curve is stored along with the correction value (Block **620**) and blocks **604-622** are executed again. If the magnitude of all the correction values is less than the threshold, however, the correction values are stored along with the sound pressure level offset.

FIG. **7** depicts a sample calibration log depicting data generated from the process described with reference to FIG. **6**. As shown in FIG. **7**, data relating to the generation of the sound pressure offset includes eleven measurements for sound levels ranging from 60.8 dB to 90.0 dB and an initial offset of 118.0. Also shown is data that includes frequency response data for three iterations of Blocks **602-624**. As depicted in FIG. **7**, after the third iteration, the FRC error (i.e., 0.4 dB) is less than 1 dB, which indicates the calibration is complete.

Referring next to FIG. **8**, shown is a block diagram **800** of one embodiment of the data collection **128**, feedback **126** and speech analysis **124** blocks depicted in FIG. **1**. While referring to FIG. **8**, simultaneous reference will be made to FIG. **9**, which is a flowchart depicting steps traversed by the speech analysis module of FIG. **8** in accordance with an exemplary embodiment. As shown in FIG. **8**, calibration data **802** is received (e.g., entered by a user or received via the network depicted in FIG. **1**) and stored in a storage device **804** that is associated with a particular microphone. An optional decoder **806** is depicted in FIG. **8**, which is configured to generate decoded calibration data in the event the calibration data is encoded.

Also shown in the storage device **804** are target settings **808** that are utilized to set values utilized in speech therapy exercises. In addition, feedback options **810** that set the type of feedback that is employed during speech therapy sessions is also depicted in the storage device **804**. In some embodiments, the target settings **808** and/or the feedback options **810** are established by a speech therapy clinician (e.g., at a remote clinical setting) and sent to the client.

As shown in FIG. **9**, when a speech therapy session is initiated, a voice signal from a user is received at the microphone **114** and is sampled by the capture module **812** (Blocks **902**, **904**). In the exemplary embodiment, a fundamental frequency module **814** receives the sampled signal in the time domain and applies an autocorrelation function to obtain the fundamental frequency of the sampled voice signal (Block **906**). In addition, the sampled voice signal is received by a sound pressure level module **816** and converted to the frequency domain by an FFT module **818** so as to generate a frequency representation of the sampled voice signal (Block **908**).

As depicted in FIGS. **8** and **9**, the frequency representation of the voice signal is then filtered by a filter **820** utilizing the calibration data so as to obtain a corrected voice signal (Block **910**). A power density of the corrected voice signal (in the frequency domain) is then calculated by a power-density module **822** (Block **912**), and a sound pressure level offset **824** (e.g., the sound pressure level offset obtained at Block **602**) is applied to the power density to obtain a sound pressure level for the sampled voice signal (Block **914**). Both the sound pressure level and the fundamental frequency of the sampled voice signal are then stored (Block **916**).

If more samples of the user's voice are desired (Block **918**), then Blocks **904** to **916** are traversed again. As depicted in FIG. **9**, the highest sound pressure level among the samples is selected as the stored sound pressure level reading and an average value of the fundamental frequency is selected as the fundamental frequency. In some embodiments for example, five separate samples (e.g., 200 ms samples) are taken over the course of one second and the highest sound pressure level among the five samples is selected as the stored sound pressure level reading and an average value of the fundamental frequency is selected as the fundamental frequency.

As shown in FIG. **8**, the sound pressure level readings and/or average values of the user's fundamental frequency may be stored in N exercise-data files (e.g., one or more data files), that may be analyzed at the client location **106** or sent to the clinic **108** for further analysis.

The exercise/feedback module **826** may be configured to provide exercises (e.g., video exercises) that prompt a user to perform particular speech exercises (e.g., vocalizing at various frequencies or speaking functional phrases) and may provide feedback (e.g., graphical or audible) to facilitate proper sampling and provide a therapeutic feedback system to help improve the speech of a user.

In many embodiments, the sound-pressure module **816** may be realized by a dynamic link library (DLL) module that may be seamlessly imported by a variety of software applications to utilize generated calibration data to substantially improve the quality sound measurements.

Although several embodiments of the present invention are described herein in the context of a speech therapy environment, it is certainly contemplated that the techniques for calibrating a transducer (e.g., a microphone) described herein have applications in a variety of contexts where sound analysis and/or feedback systems, outside of the therapy environment, is useful.

In conclusion, the present invention provides, among other things, a system and method for accurately measuring sound. Those skilled in the art can readily recognize that numerous variations and substitutions may be made in the invention, its use and its configuration to achieve substantially the same results as achieved by the embodiments described herein. Accordingly, there is no intention to limit the invention to the disclosed exemplary forms. Many variations, modifications and alternative constructions fall within the scope and spirit of the disclosed invention as expressed in the claims.

What is claimed is:

1. A method for measuring sound comprising:

sequentially generating frequency-banded-noise samples, wherein each frequency-banded-noise sample corresponds to a frequency band, each frequency banded noise sample being generated at a different point in time, wherein the frequency bands collectively cover at least a portion of a spectrum;

receiving a baseline sound-pressure-level reading from a sound level meter for each of the frequency banded noise samples;

generating, utilizing data received from a microphone, a corresponding sound pressure level reading for each of the same frequency banded noise samples the baseline sound-pressure-level reading is received for from the sound level meter;

producing calibration data for the microphone as a function of a difference between each of the baseline sound-pressure-level readings and a corresponding one of each of the generated sound pressure level readings for each of the frequency banded noise samples; and

providing the calibration data and the microphone to a user.

2. The method of claim **1**, wherein the providing includes providing the calibration data and the microphone simultaneously to a user.

3. The method of claim **1**, wherein the providing includes providing the calibration data to the user subsequent to the user receiving the microphone.

4. The method of claim **3**, wherein the providing includes providing the calibration data to the user via the Internet.

5. The method of claim **1**, including:

encoding the calibration data so as to generate encoded calibration data for the microphone.

6. The method of claim **1**, wherein each of the frequency-banded-noise samples spans an octave band of a spectrum of noise.

7. The method of claim **6**, wherein each of the frequency-banded-noise samples spans an octave band of a spectrum of pink noise.

8. The method of claim **1**, including:

generating an offset for the microphone based upon a difference between a power-density reading of a noise sample received from the microphone with a sound-pressure reading of the noise sample received from the sound-level meter;

wherein producing calibration data includes producing the calibration data so as to include the offset.

9. A system for measuring sound including a processor in communication with a memory that controls operation of:

a first input configured to receive a sound-pressure-level reading from a sound level meter for each of a plurality of frequency-banded-noise samples, each of the frequency-banded-noise samples corresponding to a frequency band, and each of the frequency banded noise samples being generated at a different point in time, wherein the frequency bands collectively cover at least a portion of a spectrum;

a second input configured to receive, from a microphone, data corresponding to each of the same plurality of frequency banded noise samples the baseline sound-pressure-level reading is received for from the sound level meter; and

a calibration module configured to generate calibration data for the microphone as a function of a difference between each of the sound-pressure-level readings received via the first input and a corresponding one of each of a plurality of generated-sound-pressure-level readings, the plurality of generated-sound-pressure-level readings being generated from the data corresponding to each of the plurality of frequency-banded-noise samples received from the second input.

10. The system of claim **9**, wherein the calibration module is configured to generate the frequency-banded-noise samples.

11. The system of claim **10**, wherein the memory includes at least one audio data file encoded with frequency-banded-noise data.

12. The system of claim **11**, wherein the memory is selected from the group consisting of a hard drive, random-access memory and read-only memory.

13. The system of claim **9**, wherein the calibration module is configured to encode the calibration data.

14. The system of claim **9**, wherein the calibration module is configured to generate an offset for the microphone based upon a difference between a power-density reading of a noise sample received from the microphone with a sound-pressure reading of the noise sample received from the sound-level meter, wherein the calibration module is configured to include the offset as a component of the calibration data.

15. A non-transitory processor-readable medium encoded with instructions for measuring sound, the instructions including instructions for:

sequentially generating frequency-banded-noise samples, wherein each frequency-banded-noise samples corresponds to a frequency band, each frequency banded noise sample being generated at a different point in time, wherein the frequency bands collectively cover at least a portion of a spectrum;

receiving a baseline sound-pressure-level reading from a sound level meter for each of the frequency banded noise samples;

generating, utilizing data received from a microphone, a sound pressure level reading for each of the same frequency banded noise samples the baseline sound-pressure-level reading is received for from the sound level meter; and

producing calibration data for the microphone as a function of a difference between each of the baseline sound-pressure-level readings and a corresponding one of each of the generated sound pressure level readings for each of the frequency banded noise samples.

16. The non-transitory processor-readable medium of claim **15**, wherein the instructions include instructions for providing the calibration data to a user, via the Internet, subsequent to the user receiving the microphone.

17. The non-transitory processor-readable medium of claim **15**, including instructions for:

encoding the calibration data so as to generate encoded calibration data for the microphone.

18. The non-transitory processor-readable medium of claim **15**, wherein each of the frequency-banded-noise samples spans an octave band of a spectrum of noise.

19. The non-transitory processor-readable medium of claim **18**, wherein each of the frequency-banded-noise samples spans an octave band of a spectrum of pink noise.

20. The non-transitory processor-readable medium of claim **15**, including instructions for:

generating an offset for the microphone based upon a difference between a power-density reading of a noise sample received from the microphone with a sound-pressure reading of the noise sample received from the sound-level meter;

wherein producing calibration data includes producing the calibration data so as to include the offset.

21. A non-transitory processor-readable medium encoded with instructions for measuring sound, the instructions including instructions for:

converting a sound sample from a microphone into a frequency domain so as to generate frequency information relative to the sound sample;

filtering the frequency information with calibration data for the microphone, the calibration data including data dependent upon a difference between each of a plurality of baseline sound-pressure-level readings received from

a sound level meter and a corresponding one of each of a plurality of generated sound pressure level readings for each of a plurality of frequency banded noise samples, wherein each of the baseline sound-pressure-level reading and corresponding generated sound pressure level reading are from a single frequency banded noise sample of the plurality of frequency banded noise samples;

generating a power density for the filtered frequency information so as to obtain a corrected sound sample; and

applying a sound-pressure level offset to the power density so as to obtain a sound-pressure level for the sound sample.

22. The non-transitory processor-readable medium of claim **21**, wherein the instructions include instructions for dynamically linking with a software application, wherein the software application is adapted to utilize the sound-pressure levels.

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